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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/891,876	06/26/2001	Shingo Kiuchi	9333/274	9050

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EXAMINER	
VO, HUYEN X	
ART UNIT	PAPER NUMBER
2655	

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/891,876	Applicant(s) KIUCHI ET AL.	
	Examiner Huyen Vo	Art Unit 2655	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *28 September 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-18* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-18* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *26 June 2001* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-3, 5-10, 12-14, and 16-17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Mokbel et al. (US 5905969) in view of Im et al. (US 5805696).

3. Regarding claim 1, Mokbel et al. disclose a voice feature extraction device comprising: a noise reduction system coefficient calculation unit that calculates adaptation filter parameters based on the LMS error between the reference signal and the input signal (*col. 5, line 61 to col. 6, line 36*), and an input voice power spectrum calculation unit that calculates a power spectrum vector of a power spectrum signal produced from an input voice signal (*figure 2, particularly elements 1000 and 1003*), wherein the noise reduction system that is set to the coefficient calculated by the noise reduction system coefficient calculation unit executes a noise reduction processing on the power spectrum vector (*element 30 of figure 3a or referring to col. 5, lines 61-67*).

Mokbel et al. fail to specifically disclose that the noise reduction system coefficient calculation unit that adds a simulated voice signal to a surrounding signal, and calculates a noise reduction system coefficient of a noise reduction system.

However, Im et al. teach that the noise reduction system coefficient calculation unit that adds a simulated voice signal to a surrounding signal, and calculates a noise reduction system coefficient of a noise reduction system (*figures 3 and 4*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate and update filter coefficients to appropriately reduce the noise corrupting the signal.

4. Regarding claim 8, Mokbel et al. disclose a voice feature extraction device comprising: a noise reduction system coefficient calculation unit that calculates adaptation filter parameters based on the LMS error between the reference signal and the input signal (*col. 5, line 61 to col. 6, line 36*), and a microphone that collects an input voice signal of a user (*a microphone is inherently needed in converting acoustic sound to electrical signal s(n) in figure 2*), a window function operation unit that samples the voice signal received by the microphone, and prevents generation of high frequency components caused by a data jump at intervals of each frame (*col. 4, lines 21-35*), an input voice power spectrum calculation unit that calculates a power spectrum vector of the input voice signal processed by the window function operation unit (*figure 2, particularly elements 1000 and 1003*), and a noise reduction system that is set to the coefficient calculated by the noise reduction system coefficient calculation unit, and

executes a noise reduction processing on the power spectrum vector (*element 30 of figure 3a or referring to col. 5, lines 61-67*).

Mokbel et al. fail to specifically disclose that the noise reduction system coefficient calculation unit that adds a simulated voice signal to a surrounding signal, and calculates a noise reduction system coefficient of a noise reduction system. However, Im et al. teach that the noise reduction system coefficient calculation unit that adds a simulated voice signal to a surrounding signal, and calculates a noise reduction system coefficient of a noise reduction system (*figures 3 and 4*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate and update filter coefficients to appropriately reduce the noise corrupting the signal.

5. Regarding claim 12, Mokbel et al. disclose a method of extracting voice features comprising: calculating a noise reduction system coefficient of a noise reduction system to be used (*col. 5, line 53 to col. 6, line 35*), and calculating a power spectrum vector of a power spectrum signal produced from an input voice signal (*figure 2, particularly elements 1000 and 1003*), wherein the noise reduction system that is set to the calculated noise reduction system coefficient executes noise reduction processing on the power spectrum vector, and extracts the voice features (*element 30 of figure 3a or referring to col. 5, lines 61-67*). Mokbel et al. fail to specifically disclose that the step of

adding a simulated voice signal to a surrounding signal. However, Im et al. teach the step of adding a simulated voice signal to a surrounding signal (*figures 3 and 4*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate the error criterion and the corresponding filter coefficients to enhance the noise suppression capability.

6. Regarding claims 2, 9, and 13, Mokbel et al. further disclose a voice feature extraction device and method as claimed in claims 1, 8, and 10, wherein the noise reduction system coefficient calculation unit includes a filter coefficient calculation unit that determines a filter coefficient of the noise reduction system to be used, and a power calculation unit that converts the filter coefficient acquired by the filter coefficient calculation unit into the power spectrum vector (*col. 7, lines 22-55*).

7. Regarding claims 3, 10, and 14, Mokbel et al. fail to specifically disclose a voice feature extraction device and method as claimed in claims 2, 9, and 12, wherein the filter coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient. However, Im et al. further teach the step of adaptively controlling a signal having the input surrounding signal and the simulated voice signal

added, and obtains a tap coefficient to thereby calculate the filter coefficient (*element 20 in figure 3, note that T(z) is the added signal*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate and update filter coefficients to appropriately reduce the noise corrupting the signal.

8. Regarding claims 5-7, the modified Mokbel et al. fail to specifically disclose a voice feature extraction device as claimed in claim 1, wherein the voice feature extraction device is applied to a voice recognition device of a vehicle navigation system, a speaker recognition device, and a loudness compensation system. However, the examiner takes official notice that the application of the noise suppression system or speech enhancement system in the voice recognition device of a vehicle navigation system, the speaker recognition device, and the loudness compensation system is well known to a person of ordinary skill in the art. The advantage of using the noise suppression system or speech enhancement system in the voice recognition device of a vehicle navigation system, the speaker recognition device, and the loudness compensation system is to remove noise before voice/speaker recognition to enhance the recognition accuracy.

9. Regarding claim 16, Mokbel et al. disclose a method of extracting voice features comprising: calculating a noise reduction system coefficient of a noise reduction system (*col. 5, line 53 to col. 6, line 35*), and sampling an input voice signal received by a microphone (*col. 4, lines 21-35*), executing processing to prevent generation of high frequency components of the input voice signal sampled (*col. 4, lines 21-35*), calculating a power spectrum vector of a power spectrum signal produced from the input voice signal that is processed to prevent generation of high frequency components (*figure 2, particularly elements 1000 and 1003*), and calculating a voice feature from the power spectrum vector via the noise reduction system that is set to the calculated noise reduction system coefficient (*element 30 of figure 3a or referring to col. 5, lines 61-67*). Mokbel et al. fail to specifically disclose that the step of adding a simulated voice signal to a surrounding signal. However, Im et al. teach the step of adding a simulated voice signal to a surrounding signal (*figures 3 and 4*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate the error criterion and the corresponding filter coefficients to enhance the noise suppression capability.

10. Regarding claim 17, Mokbel et al. further disclose a method of extracting voice features as claimed in claim 16, wherein the noise reduction system coefficient is attained by: receiving an input voice signal and executing an adaptive control to the

received signal (*figures 3a and 3c*) and applying a fast Fourier transform to the filter coefficient to thereby calculate the power spectrum vector (*col. 7, lines 22-55*). Mokbel et al. fail to specifically disclose the step of executing an adaptive control to the added signal to thereby calculate a filter coefficient. However, Im et al. further teach the step of executing an adaptive control to the added signal to thereby calculate a filter coefficient (*element 20 in figure 4*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate and update filter coefficients to appropriately reduce the noise corrupting the signal.

11. Claims 4, 11, 15, and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Mokbel et al. (US 5905969) in view of Im et al. (US 5805696), and further in view of Haykin et al. (US 5027123).

12. Regarding claim 18, Mokbel et al. disclose a voice feature extraction device comprising: a microphone that receives a surrounding signal (*inherently included in the system*); a noise reduction system coefficient calculation unit that calculates adaptation filter parameters based on the LMS error between the reference signal and the input signal (*col. 5, line 61 to col. 6, line 36*); an FFT operation unit that executes a fast Fourier transform on the filter coefficient obtained by the adaptive control of the adaptive

filter (*col. 7, lines 22-55*); a power calculation unit that calculates a power spectrum vector from a power spectrum signal calculated by the FFT operation unit (*figure 2, particularly elements 1000 and 1003*); and a noise reduction system having the power spectrum vector calculated by the power calculation unit set as a noise reduction coefficient (*element 30 of figure 3a or referring to col. 5, lines 61-67*).

Mokbel et al. fail to specifically disclose a simulated voice signal generation unit that generates a simulated voice signal; a gain adjustment unit that adjusts a gain of the simulated voice signal; an adder that adds the surrounding signal collected by the microphone and the gain-adjusted simulated voice signal; a delay processing unit that delays the gain-adjusted simulated voice signal by a predetermined time; an adaptive filter that executes an adaptive control on the basis of the signal added by the adder and the simulated voice signal delayed by the delay processing unit, and generates a filter coefficient.

Im et al. teach a device comprising: a simulated voice signal generation unit that generates a simulated voice signal (*Training Sequence Generator 34 in figure 3*); an adder that adds the surrounding signal collected by the microphone and the gain-adjusted simulated voice signal (*Adder 30 in figure 3*); an adaptive filter that executes an adaptive control on the basis of the signal added by the adder and the simulated voice signal delayed by the delay processing unit, and generates a filter coefficient (*element 20 and 28 figure 3*).

Since Mokbel et al. and Im et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at

the time of invention to modify Mokbel et al. by incorporating the teaching of Im et al. in order to calculate and update filter coefficients to appropriately reduce the noise corrupting the signal.

The modified Mokbel et al. still fail to specifically disclose a gain adjustment unit that adjusts a gain of the simulated voice signal, and a delay-processing unit that delays the gain-adjusted simulated voice signal by a predetermined time. However, Haykin et al. teach a gain adjustment unit that adjusts a gain of the simulated voice signal (*element 33 of figure 4*), and a delay-processing unit that delays the gain-adjusted simulated voice signal by a predetermined time (*element 27, 31, 33, 35, and 37 of figure 4 together forms a delay unit*).

Since the modified Mokbel et al. and Haykin et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Mokbel et al. by incorporating the teaching of Haykin et al. in order to control the simulated voice signal to achieve a precise calculation of filter coefficients.

13. Regarding claims 4, 11, and 15, the modified Mokbel et al. further disclose a voice feature extraction device as claimed in claims 3, 9, and 14, wherein the filter coefficient calculation unit executes an adaptive control to a signal having the input voice signal and simulated voice signal added, and determines a tap coefficient to thereby calculate the filter coefficient (*see figure 3 in Im et al.*). The modified Mokbel et al. fail to specifically disclose a specific gain adjustment on the simulated voice signal.

However, Haykin et al. a specific gain adjustment on the simulated voice signal (*element 33 of figure 4*).

Since the modified Mokbel et al. and Haykin et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Mokbel et.al. by incorporating the teaching of Haykin et al. in order to control the simulated voice signal to achieve a precise calculation of filter coefficients.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Huyen Vo whose telephone number is 703-305-8665. The examiner can normally be reached on M-F, 9-5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Doris To can be reached on 703-305-4827. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Examiner Huyen X. Vo

January 3, 2005

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SUSAN MCFADDEN
PRIMARY EXAMINER